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INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

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Applicant

KONINKLIJKE PHILIPS ELECTRONICS N.V.

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 3 sheets.

☒ It is also accompanied by a copy of each prior art document cited in this report.

1. Basis of the report

a. With regard to the **language**, the international search was carried out on the basis of the international application in the language in which it was filed, unless otherwise indicated under this item.

☐ the international search was carried out on the basis of a translation of the international application furnished to this Authority (Rule 23.1(b)).

b. With regard to any **nucleotide and/or amino acid sequence** disclosed in the international application, the international search was carried out on the basis of the sequence listing :

☐ contained in the international application in written form.

☐ filed together with the international application in computer readable form.

☐ furnished subsequently to this Authority in written form.

☐ furnished subsequently to this Authority in computer readable form.

☐ the statement that the subsequently furnished written sequence listing does not go beyond the disclosure in the international application as filed has been furnished.

☐ the statement that the information recorded in computer readable form is identical to the written sequence listing has been furnished

2. ☐ **Certain claims were found unsearchable** (See Box I).

3. ☐ **Unity of invention is lacking** (see Box II).

4. With regard to the **title**,

☐ the text is approved as submitted by the applicant.

☒ the text has been established by this Authority to read as follows:

MIMO TRANSMISSION SYSTEM IN A RADIO COMMUNICATIONS NETWORK

5. With regard to the **abstract**,

☒ the text is approved as submitted by the applicant.

☐ the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this international search report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is Figure No.

☒ as suggested by the applicant.

☐ because the applicant failed to suggest a figure.

☐ because this figure better characterizes the invention.

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☐ None of the figures.

MIMO TRANSMISSION SYSTEM IN RADIO COMMUNICATIONS NETWORK

The invention relates to a transmission system for transmitting an information signal via a plurality of subchannels from a transmitter to a receiver, the transmitter comprising a demultiplexer for demultiplexing the information signal into a plurality of information subsignals, the transmitter further comprising a channel encoder for encoding the information subsignals into encoded information subsignals, wherein each encoded information subsignal is transmitted via one of the subchannels to the receiver, the receiver comprising a channel decoder for successively decoding the received encoded information subsignals.

The invention further relates to a transmitter for transmitting an information signal via a plurality of subchannels to a receiver, to a receiver for receiving encoded information subsignals via a plurality of subchannels from a transmitter, to a method of transmitting an information signal via a plurality of subchannels to a receiver and to a method of receiving encoded information subsignals via a plurality of subchannels from a transmitter.

Such a transmission system is known from European Patent Application EP 0 951 091 A2. In this known transmission system multiple transmit antennas are used to transmit streams of coded symbols (i.e. the encoded information subsignals) that originate from the same data source (i.e. the information signal). At the receiver, these multiple streams are received by means of multiple receive antennas and decoded successively by removing the data streams that have been decoded at earlier stages and by cancelling the remaining data streams, due to the multiple receive antennas, via spatial (space-time or space-frequency) interference cancellation. Such a scheme is often referred to as an ordered successive interference cancellation (OSIC) scheme.

The capacity (throughput) of the known transmission system is limited.

It is an object of the invention to provide a transmission system according to the preamble having a higher transmission capacity than the known transmission system.

This object is achieved in the transmission system according to the invention, said transmission system being characterized in that the demultiplexer is arranged for demultiplexing the information signal into a plurality of partly overlapping information subsignals comprising shared and non-shared information elements, wherein for each

5 subsignal a distribution of the shared information elements is adapted to the throughput of the subchannels as ordered by the receiver, the channel decoder being arranged for decoding a received encoded information subsignal by incorporating shared information elements of already decoded information subsignals. Consider a setup of the known transmission system with two subchannels. Assume that two data streams are transmitted over two subchannels

10 with the respective capacities C_1 and C_2 (measured in user bits or information bits per channel use), and these capacities C_1 and C_2 are known to the transmitter. Assume, however, that the association between the transmitted streams and the subchannel is unknown to the transmitter. At the receiver, the data streams may be decoded successively in an arbitrary order, but not simultaneously. Clearly, the maximum achievable capacity of the

15 known system is upper bounded by $(C_1 + C_2)$. A straightforward approach is to use two independent sets of user bits to encode both data streams. Since the association between the data streams and the subchannels is unknown to the transmitter, the bit rate of each data stream is limited by the minimum of the capacities: $\min(C_1, C_2)$. If a higher bit rate were to be used, one would risk losing data. The maximum cumulative bit rate over two

20 subchannels is therefore $2 \min(C_1, C_2)$. This rate is smaller than the upper bound $(C_1 + C_2)$ by a factor of $|C_1 - C_2|$.

Assume, without loss of generality, that $C_1 \geq C_2$. The invention is based upon the recognition that the full capacity $(C_1 + C_2)$ can be reached by encoding the data streams with two intersecting sets of user bits of equal size. Specifically, the numbers of independent

25 bits (non-shared information elements) and common bits (shared information elements) within each stream can be selected so that both streams have an equal bit rate C_1 whereas the conditional bit rate of a first stream given the second stream (i.e. only the independent/unknown bits of the first stream) is equal to C_2 . In other words, the total number of user bits encoding a data block for each stream is $N C_1$ and the number of common bits within the two

30 blocks is $N(C_1 - C_2)$, where N is the number of channel uses (or signal dimensions) involved. This principle is illustrated in Figure 1 where the sets of user bits encoding two data streams are depicted by two ovals. Each set of user bits defines a codeword of

N channel symbols, from a code that matches the channel properties. For instance, a Gaussian code should be (ideally) used for an additive white Gaussian noise (AWGN) channel. Note that the total number of user bits transmitted over two subchannels is $N(C_1 + C_2)$, hence the overall bit rate is $(C_1 + C_2)$.

5 At the receiver, these data streams are decoded successively, starting with the best subchannel, in our example, the subchannel with capacity C_1 . Clearly, decoding of the best subchannel yields all NC_1 user bits. Due to the intersection between the two sets of user bits, the remaining number of unknown user bits of the second (worst) subchannel is NC_2 . Hence the conditional bit rate over the second subchannel, after the decoding of the first
10 subchannel, is C_2 . Since this rate is equal to the capacity of the second subchannel, the remaining NC_2 user bits of the second data stream can be decoded.

 In an embodiment of the transmission system according to the invention the demultiplexer is arranged for demultiplexing the information signal into the plurality of partly overlapping information subsignals according to the user bit partitioning algorithm.
15 This algorithm ensures that the conditional rates of the subsequently decoded subsignals, given the previously decoded subsignals, are matched to the throughputs of subchannels as ordered by the receiver.

 In another embodiment of the transmission system according to the invention the transmitter further comprises an interleaver coupled between the demultiplexer and the
20 channel encoder, the interleaver being arranged for interleaving the information subsignals, wherein the channel encoder is arranged for encoding the interleaved information subsignals into the encoded information subsignals. This interleaver is meant to homogeneously spread the shared information elements within the stream of all input information elements, so as to efficiently use the shared information elements known from the previously decoded streams
25 for the decoding of the current stream.

 In a further embodiment of the transmission system according to the invention the channel decoder is arranged for decoding a received encoded information subsignal by incorporating shared information elements of the most recently decoded information subsignal. In such a case, the most recent update on a shared information element may
30 incorporate all the available estimates (reliability measures) on the information elements shared through the previously decoded substreams, thereby providing the highest reliabilities of these information elements.

The above object and features of the present invention will be more apparent from the following description of the preferred embodiments with reference to the drawings, wherein:

5 Figure 1 shows a diagram of two intersecting sets of user bits which illustrate the concept of partly overlapping information subsignals 23,

 Figure 2 shows a block diagram of a transmission system 10 according to the invention,

 Figures 3 and 5 show block diagrams of embodiments of a transmitter 12
10 according to the invention,

 Figure 4 shows a block diagram of a prior art transmitter 12,

 Figure 6 shows a block diagram of a prior art wireless receiver 16,

 Figure 7 shows a block diagram of an embodiment of a receiver 16 according to the invention,

15 Figures 8 to 13 show some graphs illustrating the performance of a transmission system 10 according to the invention.

 In the Figures, identical parts are provided with the same reference numbers.

20 This invention relates to a transmission system 10 as shown in Figure 2 which transmission system 10 makes use of multiple subchannels to deliver information from a transmitter 12 to a receiver 16. The signals from different subchannels are decoded successively. The order of decoding is defined at the receiver 16 and is unknown to the transmitter 12. It is assumed that the transmitter 12 is aware of the ordering principle and of
25 some properties of the ordered sequence of subchannels such as statistics (e.g. Rayleigh fading) of the capacities (maximum throughputs) of the ordered subchannels. Alternatively, the transmitter 12 may be aware of the signal to noise (and interference) ratio (SNR/SINR) of the ordered subchannels. However, the order of decoding for each realisation of subchannels is random, as seen by the transmitter 12. The present invention is related to a channel coding
30 strategy that makes use of the known properties of the subchannels to enhance the performance of such a multi-channel transmission system 10. This channel coding strategy comprises an encoder and a decoder architecture. The general coding strategy is further applied to enhance the performance of wireless transmission systems that make use of multiple transmit antennas to transmit multiple parallel data streams and multiple receive

antennas with an ordered extraction of the transmitted streams at the receiver known as ordered successive interference cancellation (OSIC).

First, we explain the basic idea of the proposed channel coding strategy. To this end, consider a setup with two subchannels. Assume that two data streams are transmitted over two subchannels with the respective capacities C_1 and C_2 (measured in user bits or information bits per channel use), and these capacities C_1 and C_2 are known to the transmitter 12. Assume, however, that the association between the transmitted streams and the subchannel is unknown to the transmitter 12. At the receiver 16, the data streams may be decoded successively in an arbitrary order, but not simultaneously. Clearly, the maximum achievable capacity of such a system is upper bounded by $(C_1 + C_2)$. A straightforward approach is to use two independent sets of user bits to encode both data streams. Since the association between the data streams and the subchannels is unknown to the transmitter 12, the bit rate of each data stream is limited by the minimum of the capacities: $\min(C_1, C_2)$. If a higher bit rate were to be used, one would risk losing data. The maximum cumulative bit rate over two subchannels is therefore $2 \min(C_1, C_2)$. This rate is smaller than the upper bound $(C_1 + C_2)$ by a factor of $|C_1 - C_2|$. The channel coding strategy described below allows for the full capacity $(C_1 + C_2)$ to be utilized.

Assume, without loss of generality, that $C_1 \geq C_2$. The data streams may be encoded with two intersecting sets of user bits of equal size. Specifically, we select the numbers of independent bits (non-shared information elements) and common bits (shared information elements) within each stream so that both streams have an equal bit rate C_1 whereas the conditional bit rate of a first stream given the second stream (i.e. only the independent/unknown bits of the first stream) is equal to C_2 . In other words, the total number of user bits encoding a data block for each stream is NC_1 and the number of common bits within the two blocks is $N(C_1 - C_2)$, where N is the number of channel uses (or signal dimensions) involved. This principle is illustrated in Figure 1 where the sets of user bits encoding two data streams are depicted by two ovals. Each set of user bits defines a codeword of N channel symbols, from a code that matches the channel properties. For instance, a Gaussian code should be (ideally) used for an additive white Gaussian noise (AWGN) channel. Note that the total number of user bits transmitted over two subchannels is $N(C_1 + C_2)$, hence the overall bit rate is $(C_1 + C_2)$.

At the receiver 16, these data streams are decoded successively, starting with the best subchannel, in our example, the subchannel with capacity C_1 . Clearly, decoding of the best subchannel yields all NC_1 user bits. Due to the intersection between the two sets of user bits, the remaining number of unknown user bits of the second (worst) subchannel is NC_2 . Hence the conditional bit rate over the second subchannel, after the decoding of the first subchannel, is C_2 . Since this rate is equal to the capacity of the second subchannel, the remaining NC_2 user bits of the second data stream can be decoded.

The general case of a transmission system 10 with an arbitrary number m of subchannels, ordered successive decoding of the corresponding data streams at the receiver 16 and a known sequence of ordered capacities is illustrated by a block diagram in Figure 2. According to this Figure 2, the set of user bits is encoded by a channel encoder 8 into m parallel streams that are transmitted over an m input m output channel 14. This channel 14 is represented by a permutation π of the m transmitted streams followed by m parallel subchannels with the respective capacities $C_1 \dots C_m$. The set of capacities is known to the receiver 16 and to the transmitter 12 whereas the permutation π is known to the receiver 16 only. The transmitter 12 treats π as a random permutation (specifically, π is assumed uniformly distributed over the set of m possible permutations). Without loss of generality, we may assume that received streams are ordered so that $C_1 \geq \dots \geq C_m$. At the receiver 16, these streams are decoded successively by m decoders 18 so that the n -th decoder 18 makes use of the knowledge of the user bits recovered by the previous $(n-1)$ decoders 18, $1 < n \leq m$. The m decoders 18 and their interconnections form a channel decoder.

In what follows, we describe the encoding/decoding strategy for the multiple input multiple output (MIMO) channel 14 in Figure 2 and specify the maximum throughput achievable with this strategy. To specify the capacity of this channel 14, we need to define a finite difference operator $\nabla(\{S_k\}_{1 \leq k \leq m}, n)$ of order $0 \leq n < m$ applied to a sequence $\{S_k\}_{1 \leq k \leq m}$ of real values:

$$\begin{aligned} \nabla(\{S_k\}_{1 \leq k \leq m}, 0) &= \{S_k\}_{1 \leq k \leq m}, & \nabla(\{S_k\}_{1 \leq k \leq m}, 1) &= \{S_k - S_{k+1}\}_{1 \leq k < m}, \\ \nabla(\{S_k\}_{1 \leq k \leq m}, n+1) &= \nabla(\nabla(\{S_k\}_{1 \leq k \leq m}, n), 1), & 0 \leq n < m. \end{aligned} \quad (1)$$

The capacity C_Σ achievable with the proposed channel coding strategy is equal to the maximum of the sum of positive values $\underline{C}_1 \dots \underline{C}_m$ that satisfy two conditions, namely $\underline{C}_k \leq C_k$, $1 \leq k \leq m$ and $\nabla(\{\underline{C}_k\}_{1 \leq k \leq m}, n) \geq 0$, $0 \leq n < m$. In other terms,

$$C_\Sigma = \max (\underline{C}_1 + \dots + \underline{C}_m \quad : \quad \underline{C}_k \leq C_k, \quad 1 \leq k \leq m, \quad \nabla(\{\underline{C}_k\}_{1 \leq k \leq m}, n) \geq 0, \quad 0 \leq n < m) \quad (2)$$

5 The channel coding strategy explained above for $m = 2$ extends to an arbitrary m . As a first step, we specify a set $\underline{C}_1 \dots \underline{C}_m$ that satisfies the two aforementioned conditions with the maximum possible sum $(\underline{C}_1 + \dots + \underline{C}_m)$. This problem can be solved by standard linear programming. For a given set of capacities $\underline{C}_1 \dots \underline{C}_m$ and the number of channel uses N , we specify m intersecting sets of information bits, $I_1 \dots I_m$, to be transmitted over m
 10 subchannels, according to the user bit partitioning (UBP) algorithm specified below. In the description of this algorithm and later throughout the document, we will make use of the following notations:

- | | | |
|----|----------------------------------|---|
| | $ \cdot $ | - size of the set (number of bits within the set); |
| | $\lfloor \cdot \rfloor$ | - lower integer bound of a non-negative value; |
| 15 | \cap | - intersection of sets; |
| | \cup | - union of sets; |
| | \subset | - inclusion relation; |
| | $\overline{(\cdot)}$ | - complementary part of the set (\cdot) e.g. elements that are not in (\cdot) ; |
| | $\overline{\{l_1, \dots, l_p\}}$ | - complement of the subset $\{l_1, \dots, l_p\} \subset \{1, \dots, n\}$ to $\{1, \dots, n\}$ |

. Compute a set $D_1 \dots D_m$ of non-negative integers: $D_n = \lfloor \nabla(\{N_{\underline{C}_k}\}_{1 \leq k \leq n}, n-1) \rfloor$, $1 \leq n \leq m$.
Choose I_1 as an arbitrary set of user bits of size $|I_1| = D_1$.

for $n = 1$ to $(m-1)$

 choose a set $A^{(n,0)} \subset \bigcap_{1 \leq l \leq n} I_l$ such that $|A^{(n,0)}| = D_n$;

 initialise $I_{n+1} := A^{(n,0)}$;

 for $p = 1$ to $(n-1)$

 for all p -tuples $\{l_1, \dots, l_p\} \subset \{1, \dots, n\}$

 choose a set $A_{l_1, \dots, l_p}^{(n,p)} \subset \left(\bigcap_{l \in \{l_1, \dots, l_p\}} \overline{I_l} \cap \bigcap_{l \in \overline{\{l_1, \dots, l_p\}}} I_l \right)$

 such that $|A_{l_1, \dots, l_p}^{(n,p)}| = \nabla(\{D_l\}_{n-p+1 \leq l \leq n+1}, p)$;

 update $I_{n+1} := I_{n+1} \cup A_{l_1, \dots, l_p}^{(n,p)}$;

 end

end

choose an arbitrary set $A^{(n,n)}$ such that $A^{(n,n)} \cap \left(\bigcup_{1 \leq l \leq n} I_l \right) = \emptyset$

and $|A^{(n,n)}| = \nabla(\{D_l\}_{1 \leq l \leq n+1}, n)$.

update $I_{n+1} := I_{n+1} \cup A^{(n,n)}$;

end

User bit partitioning algorithm

Each set I_k may be interpreted as a set of indexes of the user bits within the data block that are transmitted over the k -th subchannel, $1 \leq k \leq m$. Hence, the sets $I_1 \dots I_m$ may be

- 5 specified off-line, given the set of capacities $C_1 \dots C_m$ that characterise the MIMO channel
14. A general block diagram of the transmitter 12 is shown in Figure 3. First, the data block to be transmitted (i.e. the user bits of the information signal 21) is partitioned/demultiplexed by a demultiplexer 20 into m streams/information subsignals 23, according to the UBP algorithm 22. As a matter of fact, each of these streams 23 contains D_1 user bits and any n
- 10 streams share D_n user bits, $1 < n \leq m$. These m streams 23 may be further encoded into sequences of channel symbols 25 with identical codes by m encoders 24; these codes should

be adapted to the channel properties. The m encoders 24 form the channel encoder 8. At the receiver 16, the encoded sequences of symbols 25 (or encoded information subsignals 25) are ordered according to the capacities $C_1 \dots C_m$ and are decoded subsequently, as indicated in Figure 2. Here all m decoders 18 may have identical structures. Furthermore, the decoder 18 of the n -th stream makes use of the known D_n user bits from the n -th input stream that are shared with previous $(n-1)$ streams, $1 < n \leq m$.

It can be proved that such a scheme with optimally chosen codes and proper decoding algorithms has a potential to reach the capacity C_Σ specified in (2). The choice of encoder/decoder for each data stream deserves a particular attention. Random codes with maximum likelihood (ML) decoding at the receiver imply the maximum throughput. However, the encoding and decoding is not feasible for these codes. Practical choice of codes and corresponding decoding algorithms depend on the desirable performance characteristics (bit and frame error rates) and complexity/storage limitations. In many cases, each subchannel appears to be a scalar channel with well-understood properties, thereby giving rise to the standard channel codes. Some examples of practical interest will now be discussed.

First, a transmission system having MIMO fading channels with additive Gaussian noise will be discussed. In this context it is assumed that the signal at the output of each subchannel is given by a linear combination of the symbols transmitted in all subchannels and an additive Gaussian noise. In such scenarios, with an appropriate processing at the receiver 16, each subchannel may be regarded as a scalar channel corrupted by a residual inter-subchannel interference and additive Gaussian noise. The commonly used coding strategy consists of a bit-space encoding (FEC encoder), mapping the coded bits into channel symbols and putting these symbols into channel (modulation). The last step depends on the channel properties. Usually, modulation is accomplished in time domain (single carrier systems) or frequency domain (multicarrier systems). In both cases, spreading may be applied so that several channel symbols share several channel uses in time or frequency domain (giving rise to direct sequence or multicarrier spread spectrum transmission, respectively). The choice of channel symbol alphabet (signalling) depends on the desired spectral efficiency and FEC rate. The commonly used signalling schemes are: BPSK, QPSK, 8-PSK and 2^k -QAM with $k \geq 2$. Note that the proposed encoding strategy may be applied for such MIMO fading channels. To this end, one has to specify the set of capacities $C_1 \dots C_m$ which are characterised by the signal to interference plus noise ratios (SINR) within the

respective subchannels. In the case of fading channels, the SINR may be unknown to the transmitter 12. The standard approach is to use outage values of SINR that are chosen, according to the expected statistical properties of the fading, as a lower bound on the actual unknown SINR for all but a small subset of outage channels. Note that the outage value of
 5 SINR/capacity/etc. is the value of SINR/capacity/etc. for which the outage rate is equal to a certain percentage of the cases/time when the actual SINR/capacity/etc. of the system is worse than the outage value.

The FEC coding strategies commonly used for channels with Gaussian noise are the standard convolutional codes and, since recently, parallel and serial concatenated
 10 interleaved (turbo) codes and low density parity check (LDPC) codes. Although all these codes may be incorporated within the encoders 24 in Figure 3, the efficiency of our approach may depend on the properties of the chosen FEC code. Indeed, the error correction capability of the known user bits at the input of the FEC encoders 24 depends on the span of typical error patterns that involve the known bits. In the case of random-like codes, such as LDPC
 15 and turbo codes, typical error patterns span a substantial part of the coded bits. Therefore, every known bit at the input of the encoders 24 is expected to have a global error correction effect over the whole code (unlike a local effect in the case of convolutional codes). It is, therefore, expected that the use of turbo codes, LDPC or similar codes FEC codes will be particularly beneficial for the general MIMO encoding scheme disclosed in this document.
 20 For such FEC codes, those user bits at the input of the n -th encoder 24 which also contribute to the inputs of some other encoders 24, may be homogeneously distributed within the stream of user bits at the input of the n -encoder 24, $1 \leq n \leq m$. Such a distribution may be achieved by an interleaver (not shown) coupled between the demultiplexer 20 and the channel encoder 8 and which interleaver performs a pseudo-random (uniform) or equidistant interleaving of
 25 each stream of user bits 23 at the output of the demultiplexer 20, prior to the FEC encoding (e.g. by turbo or LDPC codes), see Figure 3. The interleaver may be further optimised subject to a chosen FEC code.

The general decoding scheme is depicted in the receiver 16 of Figure 2. It implies that at the n -th decoding stage, the n -th decoder 18 makes use of the known input
 30 bits that are shared with codes decoded at the previous $(n-1)$ stages. A practical implementation of this idea depends on the type of code and the corresponding decoding procedure. Here we distinguish between soft and hard decision decoders. In hard decision decoding, a FEC decoder generates binary decisions on the input user bits. These decisions

on shared bits (information elements) are used, at the subsequent decoding stages, to limit the choice of possible codewords.

Example: for convolutional codes, binary decisions on the input bits usually result from a ML sequence detection by Viterbi algorithm. The decisions on the shared bits are then incorporated in the Viterbi algorithm of the successive decoding stages so that, at every trellis section associated to the previously decoded shared bit, only those state transitions are considered that correspond to the decoded binary value (thereby reducing the total number of possible transitions by 2 if the conventional Viterbi algorithm is applied).

In soft decision decoding, a FEC decoder generates (soft) real-valued metrics that represent reliability measures of the input bits. Usually, each soft metric is (an approximate) log-likelihood ratio, i.e. the logarithm of the ratio of a posteriori probability of an input bit being 0 versus a posteriori probability of this bit being 1, given the observed signal. Soft metrics are often involved in decoding procedures of concatenated codes and iterative decoding algorithms. The final decision on the input bit is taken according to the sign of such soft metrics. Whenever soft decisions on the input bits are available, the use of soft metrics on the shared bits at the subsequent decoding stages, instead of binary decisions (or equivalently, very big soft values) ensures generally a better performance.

Example: soft decision decoding is commonly used to iteratively decode turbo codes and LDPC codes. For turbo codes, the soft metrics of the input bits are produced by soft input soft output (SISO) decoders of components codes. For LDPC codes, the soft metrics result from the so-called message passing decoding algorithm. Whenever such FEC codes are used within the MIMO encoding scheme disclosed in this document, the soft metrics of the shared bits, obtained at the last iteration of the current stage, are transferred to the subsequent decoding stages. At the subsequent stages, the so-obtained soft metrics should be used as a priori metrics on the shared bits or added to the existing a priori metrics if the latter are available at these decoding stages.

Finally, at every decoding stage, whenever a given input bit is shared with more than one of previous stages, the decoder preferably makes use of (either hard or soft) decisions that come from the most recent previous decoding, i.e. the channel decoder is preferably arranged for decoding a received encoded information subsignal by incorporating shared information elements of the most recently decoded information subsignal.

Iterative decoding may be applied to improve the performance of the proposed MIMO scheme. Iterative decoding of the scheme in Figure 2 means repeating the whole or any part of the full decoding cycle which consists of the successive decoding of m

subchannels as described above. In this case, the early decoding stages can make use of hard/soft information on the shared bits obtained at the subsequent stages during the previous iterations. In the case of soft decision decoding, the reliability values to be used at other stages should be computed according to the standard rules for extrinsic information, similarly to turbo decoding, in order to avoid double-counting of the same information.

The same channel coding principle may be applied to a binary MIMO transmission channel. A possible scenario is transmission of binary messages (data packets) in a network from a transmitter to a receiver via multiple routes (subchannels). Assume that exact reliabilities of different paths (crossover probabilities of the equivalent binary symmetric channels) are not known to the transmitter but the statistical properties (a distribution law of the crossover probability) are known, e.g. from modelling results. One way to deal with the uncertainty of reliabilities per subchannel is to decode these streams in such an order that the reliabilities of the ordered subchannels form a non-increasing sequence. As a matter of fact, variations of each element of an ordered sequence of independent identically distributed random values go to zero as the sequence length goes to infinity. Hence, one can accurately adapt transmission rates to the (quasi-deterministic) throughputs of the ordered subchannels provided that their number is big enough. Although the throughputs of the ordered subchannels may be assumed accurately known at the transmitter, the order of the extraction of the transmitted streams is unknown (unless a feedback channel between the transmitter and the receiver is used to convey channel information). In such a case, the overall MIMO channel falls under the scope of the general scheme of Figure 2. Further implementations may be based on the available FEC schemes for binary symmetric channels (i.e. with MDS codes). Yet, it is desirable to adapt the existing coding schemes so as to emphasise the benefit of a partial knowledge of the input bits, according to the channel coding strategy proposed in this document.

The channel coding strategy described above may also be applied to increase the throughput of wireless communication systems that exploit multiple transmit and receive antennas. In such systems, multiple transmit antennas are used to transmit streams of coded symbols that originate from the same data source. At the receiver side, these multiple streams are retrieved and decoded either simultaneously or successively. Simultaneous decoding of different streams yields a very high computational burden which grows exponentially in the total number of bits per channel use transmitted by all antennas. Therefore, simultaneous decoding is feasible only for small data rates as compared to the theoretical throughput. Here, we focus on the successive decoding schemes where each data

stream is recovered by removing the data streams that have been recovered at earlier stages and by cancelling the remaining data streams, due to multiple receive antennas 40, via spatial (space-time or space-frequency) interference cancellation. Specifically, we consider schemes with ordered successive interference cancellation (OSIC).

5 A baseline system that makes use of the OSIC principle is disclosed in European Patent Application EP 0 951 091 A2. According to this known system, the total number of user bits 21 is demultiplexed into m symmetric streams 23. Each stream 23 undergoes an identical encoding (by encoders 24), modulation (by modulators 26) and is transmitted by one of m transmit antennas 28. A block diagram of such a transmitter 12 is
 10 shown in Figure 4. The receiver 16 exploits M antennas 40 that produce M signal outputs. The receiver 16 applies the OSIC principle which is schematically shown in Figure 6. It is assumed that the transfer function of the MIMO channel is known or accurately estimated at the receiver (e.g. due to the standard training procedure, based on the reference signals sent by the transmitter). This MIMO transfer function will be schematically described by
 15 an $M \times m$ matrix H whose entry $H_{q,p}$ stands for the transfer function between the p -th transmit antenna 28 and the q -th receive antenna 40. In frequency selective fading, the entries of H are functions representing either time domain or frequency domain characterisation of the channel. In non-selective (flat) fading environments, H has complex-valued entries.

20 The received encoded information subsignals are demodulated in demodulators 42. Based on the known H , the receiver 16 extracts the m streams (information subsignals) successively from the demodulated streams 41. At the first (leftmost) layer or stage (comprised of a MMSE canceller 44, a decoder 46, an encoder 48, multipliers 56 and subtractors 60), one of the streams 41 is extracted, by cancelling the
 25 contributions from the other $(m-1)$ streams 41. Without loss of generality, assume that the index of the stream extracted at the first layer is $\pi[1]$. In the known system, a perfect cancellation of these streams is achieved due to projecting the vector

$H_{\pi[1]} = [H_{1,\pi[1]}, \dots, H_{M,\pi[1]}]^T$ (the superscript (T) stands for a matrix transpose) of channel transfer functions associated to this stream onto the part of the M -dimensional signal space
 30 which is orthogonal to the columns of the $M \times (m-1)$ matrix $[H_{1:M,\pi[2]}, \dots, H_{1:M,\pi[m]}]$ representing the transfer functions of the other streams. The $\pi[1]$ -th stream is a result of linear combining of signals from M antennas 40 with weights that are defined by the entries

of the projected vector. This type of interference cancellation, known as zero-forcing, remains sub-optimal in the presence of noise. A better performance may be achieved with minimum mean square error (MMSE) cancellation (in MMSE canceller 44) which maximises the SINR. To apply MMSE extraction to the $\pi[1]$ -th stream, we compute the $m \times M$ vector

$$W_{\pi[1]}^{(1)} = \left(H_{\pi[1]}^* \left(\sigma_s^2 H H^* + \sigma_n^2 I_M \right)^{-1} H_{\pi[1]} \right)^{-1} H_{\pi[1]}^* \left(\sigma_s^2 H H^* + \sigma_n^2 I_M \right)^{-1}, \quad (3)$$

where the superscript (*) stands for a matrix conjugate transpose, I_M is the $M \times M$ identity matrix, σ_s^2 is the (average) power of every transmitted signal and σ_n^2 is the ambient noise power. The $\pi[1]$ -th stream is a result of linear combining of signals from M antennas with weights that are defined by the respective entries of $W_{\pi[1]}^{(1)}$. Whenever the ambient noise at different antennas is uncorrelated, MMSE cancellation results in the highest possible SINR:

$$SINR_{\pi[1]}^{(1)} = \left[\left(H_{\pi[1]}^* \left(\sigma_s^2 H H^* + \sigma_n^2 I_M \right)^{-1} H_{\pi[1]} \right)^{-1} - 1 \right]^{-1}. \quad (4)$$

The extracted $\pi[1]$ -th stream is forwarded by the MMSE canceller 44 to the decoder which recovers the corresponding stream of user bits. These user bits are encoded again by an encoder 48 to the sequence of channel symbols which is scaled by the respective entries of the transfer function $H_{\pi[1]}$ (by means of multipliers 56) to produce the contributions of the $\pi[1]$ -th stream to all M receiver branches. These contributions are extracted from the corresponding received signals by means of subtractors 60 as indicated in Figure 6. The resulting M signals are free of the contributions of the $\pi[1]$ -th stream.

The described procedure is applied recursively so that at the n -th layer/stage, a $\pi[n]$ -th stream is extracted with certain $SINR_{\pi[n]}^{(n)}$, after the MMSE cancellation of the remaining $(n-1)$ interfering streams by means of filter $W_{\pi[n]}^{(n)}$, its contribution is reconstructed and removed from the received signals, $1 \leq n \leq m$ (except for the last layer where the removal is not necessary). In Figure 6 also the second layer/stage and the m -th layer/stage are shown.

This second stage is comprised of MMSE canceller 50, decoder 52, encoder 54, multipliers 64 and subtractors 68. The m -th stage is only comprised of MMSE canceller 74 and decoder 76. The receiver 16 further comprises a multiplexer 72 which multiplexes the decoded

information subsignals 47 of the m layers/stages into an information signal 77 comprising user bits.

The throughput of this MIMO transmission system depends on the set of SINR values $SINR_{\pi[1]}^{(1)}, \dots, SINR_{\pi[m]}^{(m)}$. Hence, the order $\pi = \{\pi[n], 1 \leq n \leq m\}$ of the processing of m

5 streams may be critical. To highlight the impact of the processing order on the system throughput, note that the symmetry of different subchannels and the absence of channel knowledge at the transmitter, yield equal transmission rates (throughputs) to be used for all subchannels. The overall throughput of such a system equals to m times the throughput of one subchannel. Finally, the throughput per subchannel is limited by the minimum of their
10 respective throughputs which is defined by $\min\{SINR_{\pi[1]}^{(1)}, \dots, SINR_{\pi[m]}^{(m)}\}$. The maximum throughput, therefore, corresponds to the maximum of $\min\{SINR_{\pi[1]}^{(1)}, \dots, SINR_{\pi[m]}^{(m)}\}$ whereas the optimal processing order is defined by such π that maximises
15 $\min\{SINR_{\pi[1]}^{(1)}, \dots, SINR_{\pi[m]}^{(m)}\}$. As shown in the above mentioned European Patent Application, the optimal processing order π is achieved when, at every stage, the subchannel is selected
that maximises the local SINR:

$$\pi[n] = \arg \max_k \{SINR_k^{(n)} : 1 \leq k \leq m, k \neq \pi[p] : 1 \leq p < n\} \quad 1 \leq n \leq m. \quad (5)$$

A MIMO transmission system as described earlier, with the transmitter 12 and receiver 16 as in Figure 4 and Figure 6 respectively, MMSE cancellation filters as specified by (3), decision statistics as defined by (4) and a processing order as defined in (5), is

20 regarded in this document as the baseline system. Let us analyse a theoretically achievable throughput of such a system. We will assume a narrow-band (non-selective) Rayleigh fading channel with fully uncorrelated transmit/receive antennas. This means that the entries of the channel matrix are statistically independent complex Gaussian variables with zero mean and variance $(1/2)$ per complex dimension. Let us first consider a system with two

25 transmit/receive antennas: $M = m = 2$. For this setup, the outage ratios $SINR_{\pi[1]}^{(1)}$ and $SINR_{\pi[2]}^{(2)}$ for both layers have been estimated from 100000 independent Monte-Carlo trials for a wide range of the overall SNR per receive antenna (i.e. ratio of the average total signal power from all transmit antennas to the noise power at any receive antenna). The empirical SINR values for the outage rates of 10% and 1% are plotted in Figure 8 (showing outage SINR per
30 layer/stage versus the overall SNR per receive antenna for outage rates 10% (upper frame) and 1% (lower frame), 2 transmit antennas, 2 receive antennas, uncorrelated Rayleigh

fading). The corresponding outage values of capacities (maximum throughputs), computed according to the standard relationship,

$$C = \log_2 (1 + SINR) \quad [bits / channel \text{ use}] \quad (6)$$

are plotted in Figure 9 (showing outage throughput per layer/stage versus the overall SNR per receive antenna for outage rates 10% (upper frame) and 1% (lower frame), 2 transmit antennas, 2 receive antennas, uncorrelated Rayleigh fading). One can see that at small and moderate SNR, the first (leftmost) layer has a bigger throughput. Such a behaviour comes from the fact that at low SNR, the additive noise contribution is dominant within the mixture of the noise and residual interference and therefore the selection of the best subchannel, available at the first layer, results in a better capacity of this layer. The residual interference becomes more important along with the increase of SNR which explains the degradation of the first layer compared to the second layer at high SNR. The advantage of the first layer over the second layer also depends on the designed outage rate. Note that the maximum throughput of the first layer is nearly 2 times as big as the throughput of the second layer in some cases of practical interest. Namely, the region of SNR around 6-8dB, outage rate 10% and less may be relevant for cellular communications in interference-limited environments, e.g. CDMA.

As explained earlier in this section, the throughput of each subchannel within the baseline system may not exceed the minimum of throughputs observed at different layers. Therefore, the maximum total throughput of the baseline system is twice the minimum of these throughputs. A 'diamond' curve of Figure 10 shows the overall throughput of the baseline (standard) system versus the overall SNR for the outage rates of 10% (upper frame) and 1% (lower frame).

At this point, we note that the overall throughput may be increased up to the sum of the throughputs C_1 and C_2 of the two layers in the region of SNR and outage rates where $C_1 \geq C_2$. Indeed, one may notice that the transmission system with a transmitter 12 as in Figure 4 and a receiver 16 as in Figure 6 is a particular case of the general transmission scheme of Figure 2 where the capacities $C_1 \dots C_m$ stand for the outage throughputs achievable at layers 1 through m respectively while the permutation π defines the processing order of the transmitted streams. The set of outage throughputs $C_1 \dots C_m$ is defined by the statistical description of the assumed propagation environment (in our

example, uncorrelated Rayleigh fading). Usually, these throughputs are measured off-line and may be assumed known to the transmitter 12 and to the receiver 16. The permutation π depends on the channel realisation. This permutation is defined at the receiver 16, subject to the estimated channel matrix, and is therefore unknown to the transmitter 12. Hence, the baseline transmission system falls under the general scheme as depicted in Figure 2 and therefore the general channel coding principle described above applies in this case. In a case of two transmit/receive antennas, the encoding may be performed as explained above. Let us design the encoder that has to operate in presence of uncorrelated Rayleigh fading at SNR of 8 dB and 10% outage rate. In this case, the throughputs achievable at layers 1 and 2 are $C_1 \approx 1.27$ and $C_2 \approx 0.81$ user bits per channel use respectively, see Figure 9. The practically achievable throughputs of this layers are, therefore, upper-bounded by C_1 and C_2 . These upper bounds are never achieved in practice since a (small) fraction of the spectral efficiency has to be sacrificed in order to satisfy the QoS requirements in terms of error rates. This fraction depends on the desired features of FEC and QoS requirements. The definition of practical throughput relates to the FEC design which is not addressed specifically in this document. Therefore, we will assume in this example an ideal FEC so that the maximum throughputs are achievable. Suppose that a block of data to be transmitted makes use of $N = 100$ channel uses. This may correspond to e.g. to a block of 100 symbols of certain alphabet serially sent over the channel. According to the description in section 1, we have to form two sets of user bits of equal sizes $NC_1 \approx 127$ so that these sets share $N(C_1 - C_2) \approx 127 - 81 = 46$ user bits. These two sets are independently encoded, modulated and transmitted via different antennas. At the receiver 16, the standard OSIC extraction of the data streams is performed, as described earlier in this section. According to the processing order π which is determined at the receiver via (5), the stream $\pi[1]$ is extracted at the first (leftmost) layer. Since the outage throughput of this layer is C_1 , the corresponding $NC_1 \approx 127$ user bits may be successfully decoded. Recall that 46 of these user bits are shared with the stream $\pi[2]$. At the second layer, the stream $\pi[2]$ is extracted. The decoder of this stream benefits from the knowledge of 46 user bits out of the total 127 bits. The remaining $NC_2 = 81$ user bits may be successfully recovered since the throughput of the second layer is $C_2 \approx 0.81$.

Note that the proposed channel coding scheme results in the overall throughput $(C_1 + C_2) \approx 2.08$ bits per channel use which is a 28% improvement as

compared to the baseline system with the overall throughput of $2C_2 \approx 1.62$ bits per channel use. The throughputs of baseline system and the proposed modification of this latter are plotted in Figure 10 for various SNR and outage rates of 10% and 1%. The improvement of the modified transmission system over the baseline systems varies from 10% to 100% and even higher, at low and moderate SNR.

In the general case of M and m , the transmitter of the modified baseline system looks like in Figure 5 wherein the Algorithm 1 is applied to generate m intersecting sets of user bits. These sets are encoded, modulated and transmitted in parallel, via m transmit antennas. The receiver of the modified baseline system is presented in Figure 7. As compared to the conventional receiver in Figure 6, the modified receiver contains a selection block 78 that controls via gates 80 the transfer of the shared user bits (the corresponding hard/soft decisions) decoded at earlier layers to the decoders of the following layers.

In the remainder, we give a slightly more complex example of user bit partitioning by means of the UBP algorithm, for a MIMO transmission system with $M = m = 3$. Under the assumption of uncorrelated Rayleigh fading, we computed the outage SINR achievable at different layers, the corresponding outage throughputs per layer and the outage overall throughputs of the baseline (standard) and modified systems, for a wide range of SNR and outage rates 10% and 1%, see Figures 11 to 13. Figure 11 shows the outage SINR per layer/stage versus the overall SNR per receive antenna for outage rates 10% (upper frame) and 1% (lower frame), the transmission system having 3 transmit antennas, 3 receive antennas and uncorrelated Rayleigh fading. Figure 12 shows the outage throughput per layer/stage versus the overall SNR per receive antenna for outage rates 10% (upper frame) and 1% (lower frame), the transmission system having 3 transmit antennas, 3 receive antennas and uncorrelated Rayleigh fading. Figure 13 shows the outage total throughput of the standard and the modified system versus the overall SNR per receive antenna for outage rates 10% (upper frame) and 1% (lower frame), the transmission system having 3 transmit antennas, 3 receive antennas and uncorrelated Rayleigh fading.

First of all, we need to find the set of maximum throughputs $\underline{C}_1, \underline{C}_2, \underline{C}_3$ that deliver the overall throughput C_2 under feasibility constraints, as indicated in (2). The resulting values $\underline{C}_1, \underline{C}_2, \underline{C}_3$ are plotted in Figure 12 by dashed lines. Note that $\underline{C}_1 = C_1$ and $\underline{C}_3 = C_3$ in all cases whereas the middle layer throughput \underline{C}_2 is often smaller than the corresponding C_2 . As earlier, we select SNR of 8 dB, 10% outage rate and $N = 100$ channel

uses per block. From Figure 12, we find the maximum throughputs per layer $C_1 \approx 1.51$, $C_2 \approx 1.33$, $C_3 \approx 0.95$ and the maximum feasible throughputs per layer $\underline{C}_1 \approx 1.51$, $\underline{C}_2 \approx 1.23$, $\underline{C}_3 \approx 0.95$. Next, we apply the Algorithm 1 to generates three sets of user bits. First of all, we compute $N\underline{C}_1 \approx 151$, $N\underline{C}_2 \approx 123$, $N\underline{C}_3 \approx 95$ and the corresponding $D_1 = N\underline{C}_1 = 151$,

5 $D_2 = (N\underline{C}_1 - N\underline{C}_2) = 28$, $D_3 = ((N\underline{C}_1 - N\underline{C}_2) - (N\underline{C}_2 - N\underline{C}_3)) = 0$. The remaining part of the Algorithm 1 may be implemented as follows:

- choose I_1 as an arbitrary set of $D_1 = 151$ user bits;
- take the first block of $D_2 = 28$ bits (bits 1 through 28) from I_1 , obtain I_2 by appending $(D_1 - D_2) = 123$ user bits that are not in I_1 ;
- 10 - take the second block of $D_2 = 28$ bits (bits 29 through 56) from I_1 (no intersection with I_2), append the second block of $D_2 = 28$ bits (bits 29 through 56) from I_2 (no intersection with I_1), append $((D_1 - D_2) - (D_2 - D_3)) = 95$ user bits that are not shared with I_1 and I_2 , to obtain I_3 .

Note that $D_3 = 0$ implies that the three sets of user bits to be built have an empty
 15 intersection. Let us make a more general observation. For any $m \geq 3$, whenever the throughputs $C_2 \dots C_{m-1}$ lie above the straight line between C_1 and C_m , we have $\underline{C}_1 = C_1$, $\underline{C}_m = C_m$, and all $\underline{C}_1 \dots \underline{C}_m$ are equally spaced along the (imaginary) straight line that connects C_1 and C_m . In this case, $D_n = 0$ for all $m \geq 3$ and therefore any three (or more) sets
 20 of user bits have empty intersections. This observation may be used to simplify the general version of the UBP algorithm, by confining the inner loop to $p \in \{(n-3), (n-2), (n-1)\}$. This simplification may be valuable for moderate and big m .

The scope of the invention is not limited to the embodiments explicitly disclosed. The invention is embodied in each new characteristic and each combination of characteristics. Any reference signs do not limit the scope of the claims. The word
 25 "comprising" does not exclude the presence of other elements or steps than those listed in a claim. Use of the word "a" or "an" preceding an element does not exclude the presence of a plurality of such elements.

CLAIMS:

1. A transmission system (10) for transmitting an information signal (21) via a plurality of subchannels from a transmitter (12) to a receiver (16), the transmitter (12) comprising a demultiplexer (20) for demultiplexing the information signal (21) into a plurality of information subsignals (23), the transmitter (12) further comprising a channel
5 encoder (8) for encoding the information subsignals (23) into encoded information subsignals (25), wherein each encoded information subsignal (25) is transmitted via one of the subchannels to the receiver (16), the receiver (16) comprising a channel decoder for successively decoding the received encoded information subsignals, characterized in that the demultiplexer (20) is arranged for demultiplexing the information signal (21) into a plurality
10 of partly overlapping information subsignals (23) comprising shared and non-shared information elements, wherein for each subsignal (23) a distribution of the shared information elements is adapted to the throughput of the subchannels as ordered by the receiver (16), the channel decoder being arranged for decoding a received encoded information subsignal by incorporating shared information elements of already decoded
15 information subsignals.
2. The transmission system (10) according to claim 1, wherein the demultiplexer (20) is arranged for demultiplexing the information signal (21) into the plurality of partly overlapping information subsignals (23) according to the user bit partitioning algorithm.
20
3. The transmission system (10) according to claim 1 or 2, wherein the transmitter (12) further comprises an interleaver coupled between the demultiplexer (20) and the channel encoder (8), the interleaver being arranged for interleaving the information subsignals (23), wherein the channel encoder (8) is arranged for encoding the interleaved
25 information subsignals into the encoded information subsignals (25).
4. The transmission system (10) according to any one of claims 1 to 3, wherein the channel decoder is arranged for decoding a received encoded information subsignal by

incorporating shared information elements of the most recently decoded information subsignal.

5. The transmission system (10) according to any one of claims 1 to 4, wherein the transmission system (10) is a binary transmission system and wherein the information subsignals comprise differently routed binary signals.

6. The transmission system (10) according to any one of claims 1 to 4, wherein the transmission system (10) is a wireless communication system, and wherein the transmitter (12) comprises a plurality of transmit antennas (28), wherein each encoded information subsignal (25) is transmitted via one of the transmit antennas (28) to the receiver (16), and wherein the receiver (16) comprises a plurality of receive antennas (40) for receiving the encoded information subsignals.

7. A transmitter (12) for transmitting an information signal (21) via a plurality of subchannels to a receiver (16), the transmitter (12) comprising a demultiplexer (20) for demultiplexing the information signal (21) into a plurality of information subsignals (23), the transmitter (12) further comprising a channel encoder (8) for encoding the information subsignals (23) into encoded information subsignals (25), wherein each encoded information subsignal (25) is transmitted via one of the subchannels to the receiver (16), characterized in that the demultiplexer (20) is arranged for demultiplexing the information signal (21) into a plurality of partly overlapping information subsignals (23) comprising shared and non-shared information elements, wherein for each subsignal (23) a distribution of the shared information elements is adapted to the throughput of the subchannels as ordered by the receiver (16).

8. The transmitter (12) according to claim 7, wherein the demultiplexer (20) is arranged for demultiplexing the information signal (21) into the plurality of partly overlapping information subsignals (23) according to the user bit partitioning algorithm.

9. The transmitter (12) according to claim 7 or 8, wherein the transmitter (12) further comprises an interleaver coupled between the demultiplexer (20) and the channel encoder (8), the interleaver being arranged for interleaving the information subsignals (23),

wherein the channel encoder (8) is arranged for encoding the interleaved information subsignals into the encoded information subsignals.

10. The transmitter (12) according to any one of claims 7 to 9, wherein the transmitter (12) comprises a plurality of transmit antennas (28), and wherein each encoded information subsignal (25) is transmitted via one of the transmit antennas (28) to the receiver (16).

11. A receiver (16) for receiving encoded information subsignals via a plurality of subchannels from a transmitter (12), the receiver (16) comprising a channel decoder for successively decoding the received encoded information subsignals, characterized in that the encoded information subsignals are partly overlapping encoded information subsignals comprising shared and non-shared information elements, wherein for each subsignal (23) a distribution of the shared information elements is adapted to the throughput of the subchannels as ordered by the receiver (16), the channel decoder being arranged for decoding a received encoded information subsignal by incorporating shared information elements of already decoded information subsignals.

12. The receiver (16) according to claim 11, wherein the channel decoder is arranged for decoding a received encoded information subsignal by incorporating shared information elements of the most recently decoded information subsignal.

13. The receiver (16) according to claim 11 or 12, wherein the receiver (16) comprises a plurality of receive antennas (40) for receiving the encoded information subsignals.

14. A method of transmitting an information signal (21) via a plurality of subchannels to a receiver (16), the method comprising:

- demultiplexing the information signal (21) into a plurality of information subsignals (23),
- encoding the information subsignals (23) into encoded information subsignals (25),
- transmitting each encoded information subsignal (25) via one of the subchannels to the receiver (16), characterized in that the information signal (21) is demultiplexed into a plurality of partly overlapping information subsignals (23) comprising shared and non-shared information elements, wherein for each subsignal (23) a distribution of the shared

information elements is adapted to the throughput of the subchannels as ordered by the receiver (16).

15. The method of transmitting according to claim 14, wherein the information
5 signal (21) is demultiplexed into the plurality of partly overlapping information subsignals (23) according to the user bit partitioning algorithm.

16. The method of transmitting according to claim 14 or 15, the method further
10 comprising interleaving the information subsignals (23), wherein the interleaved information subsignals are encoded into the encoded information subsignals (25).

17. The method of transmitting according to any one of claims 14 to 16, wherein
the encoded information subsignals (25) are transmitted via a plurality of transmit antennas
(28) to the receiver (16).

18. A method of receiving encoded information subsignals via a plurality of
subchannels from a transmitter (12), the method comprising:
- successively decoding the received encoded information subsignals,
characterized in that the encoded information subsignals are partly overlapping encoded
20 information subsignals comprising shared and non-shared information elements, a received
encoded information subsignal being decoded by incorporating shared information elements
of already decoded information subsignals, wherein for each subsignal (23) a distribution of
the shared information elements is adapted to the throughput of the subchannels as ordered
by the receiver (16).

19. The method of receiving according to claim 18, wherein a received encoded
information subsignal is encoded by incorporating shared information elements of the most
recently decoded information subsignal.

20. The method of receiving according to claim 18 or 19, wherein the encoded
30 information subsignals are received via a plurality of receive antennas (40).

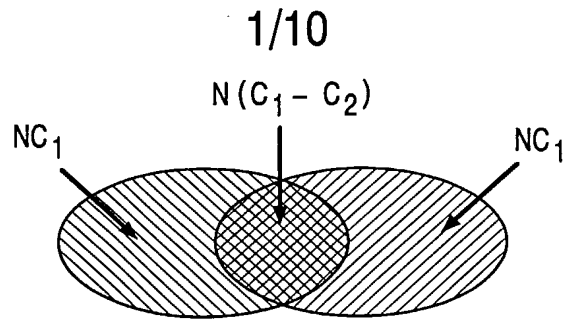


FIG. 1

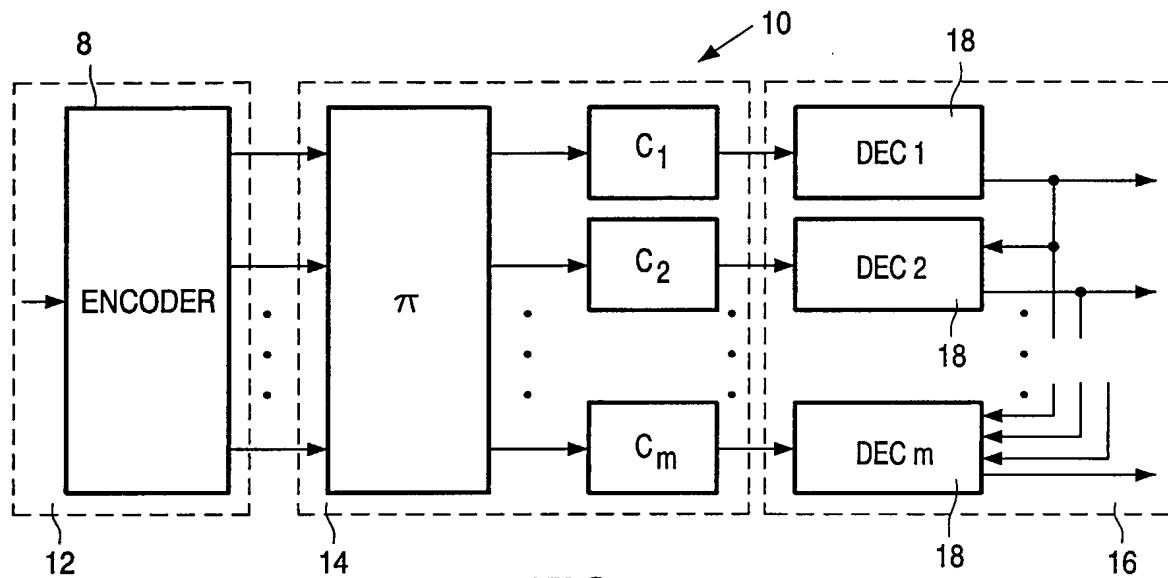


FIG. 2

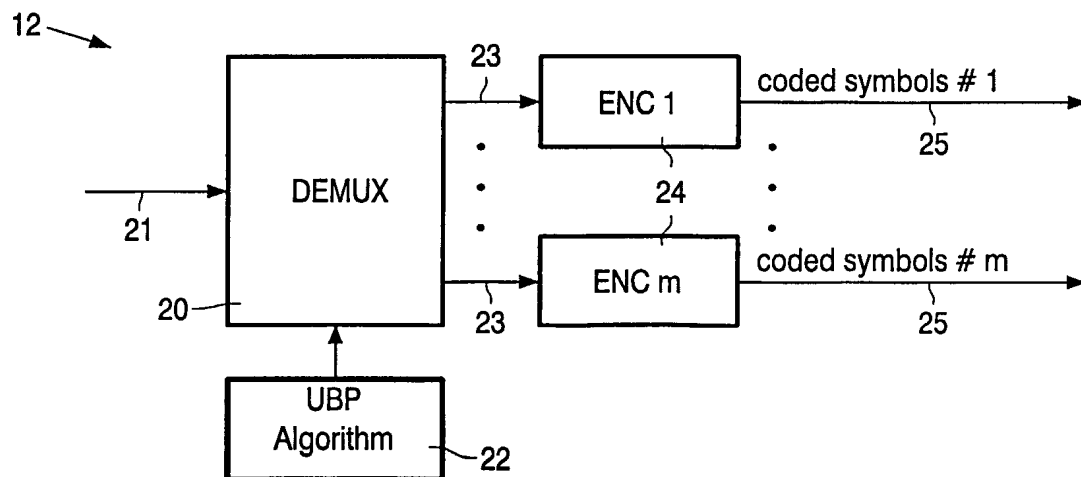


FIG. 3

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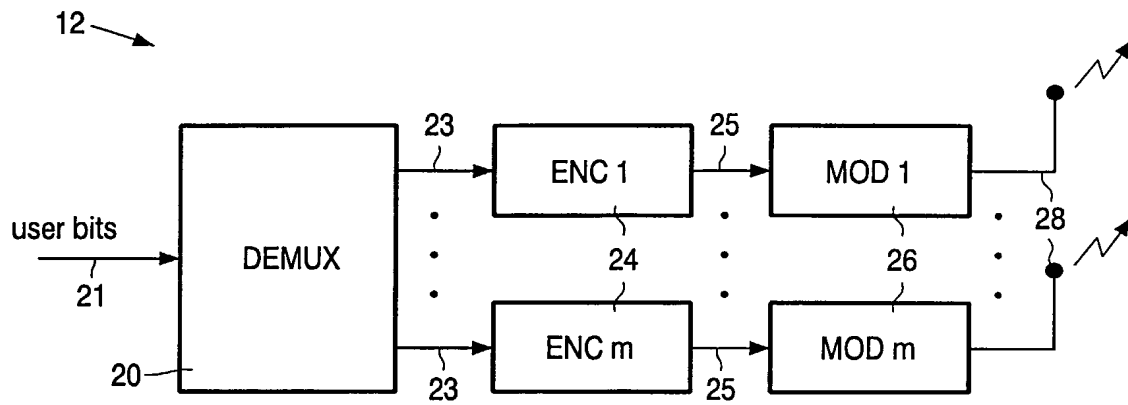


FIG. 4

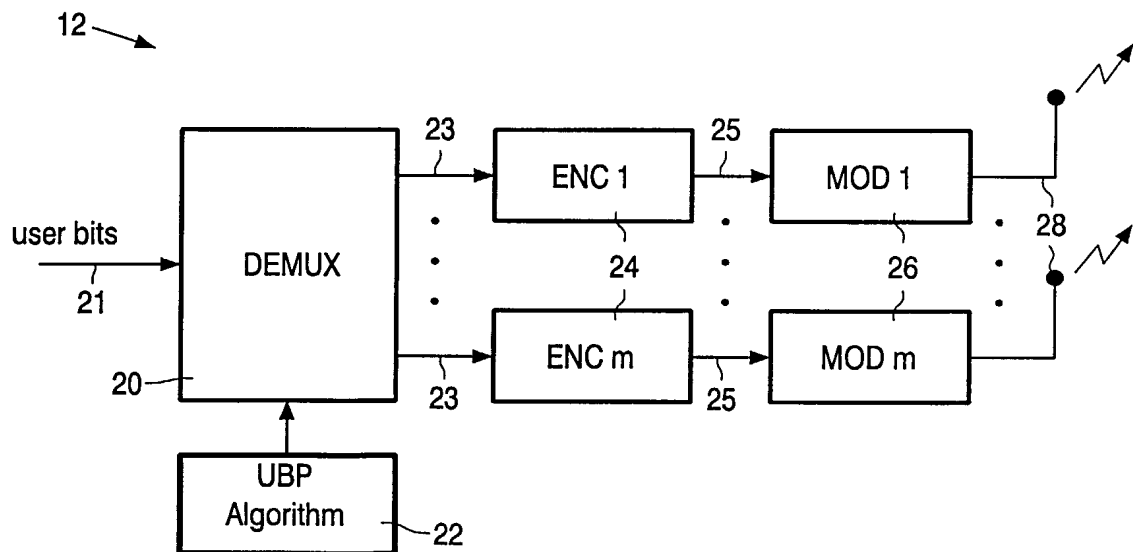


FIG. 5

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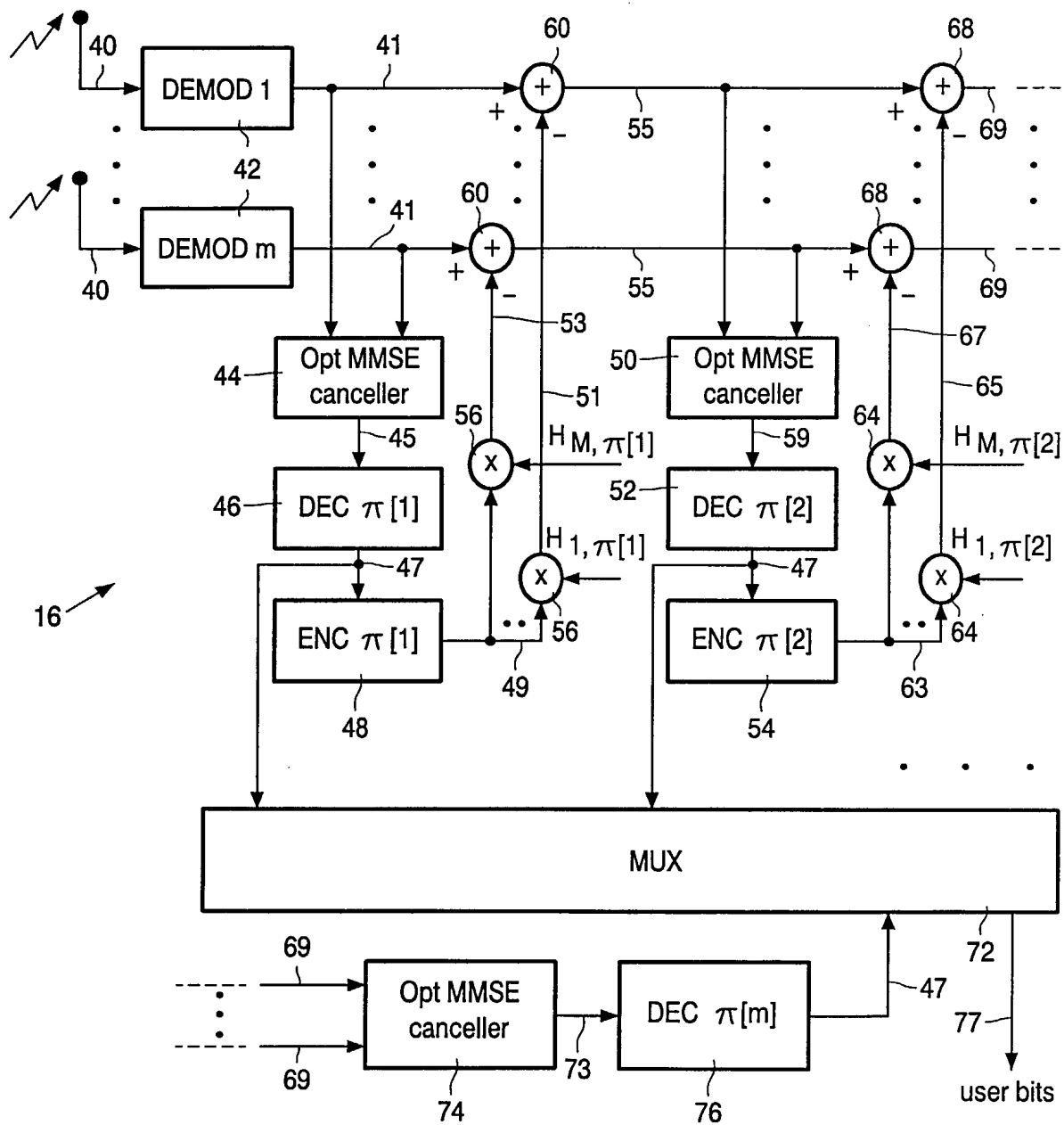


FIG. 6

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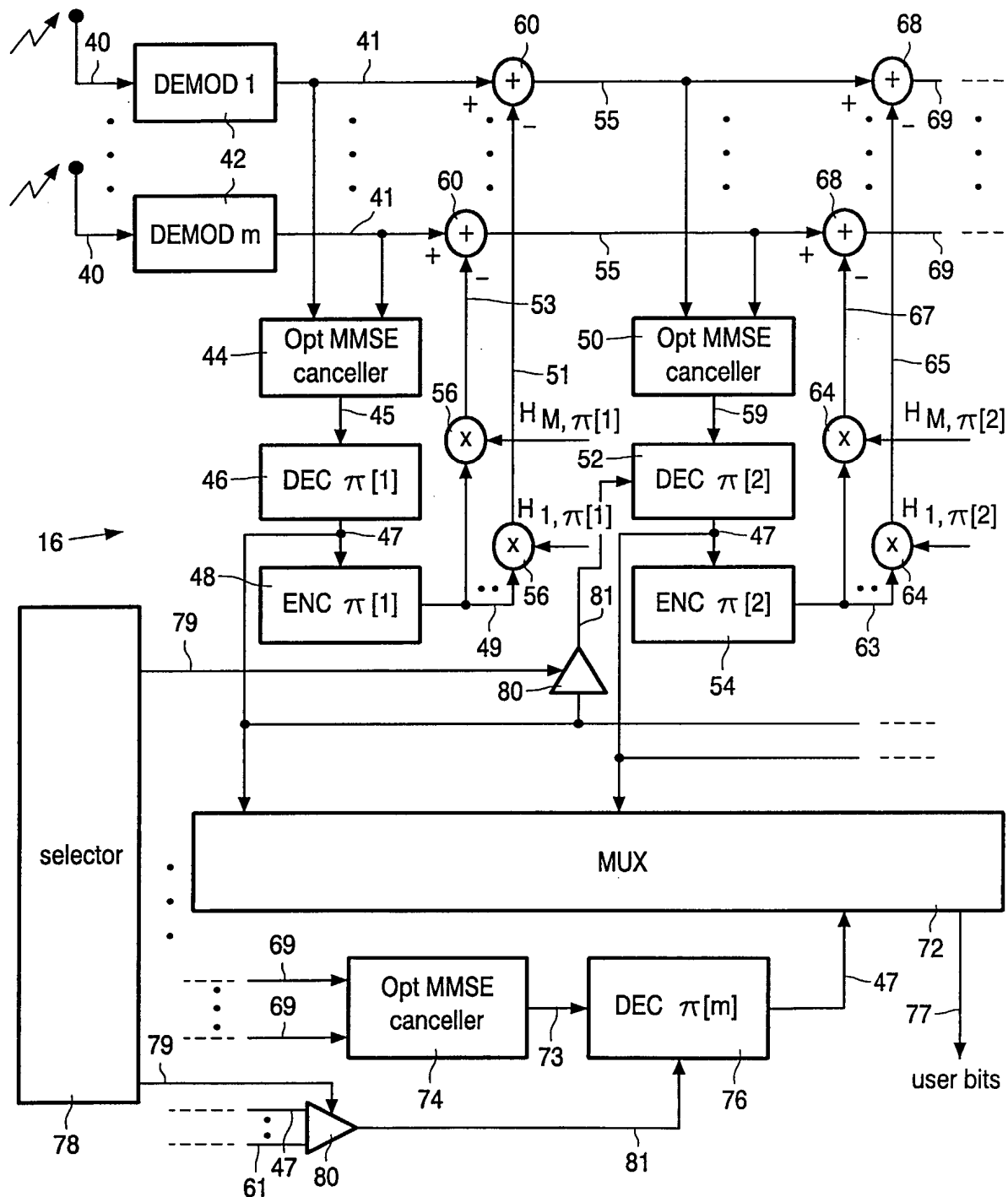


FIG. 7

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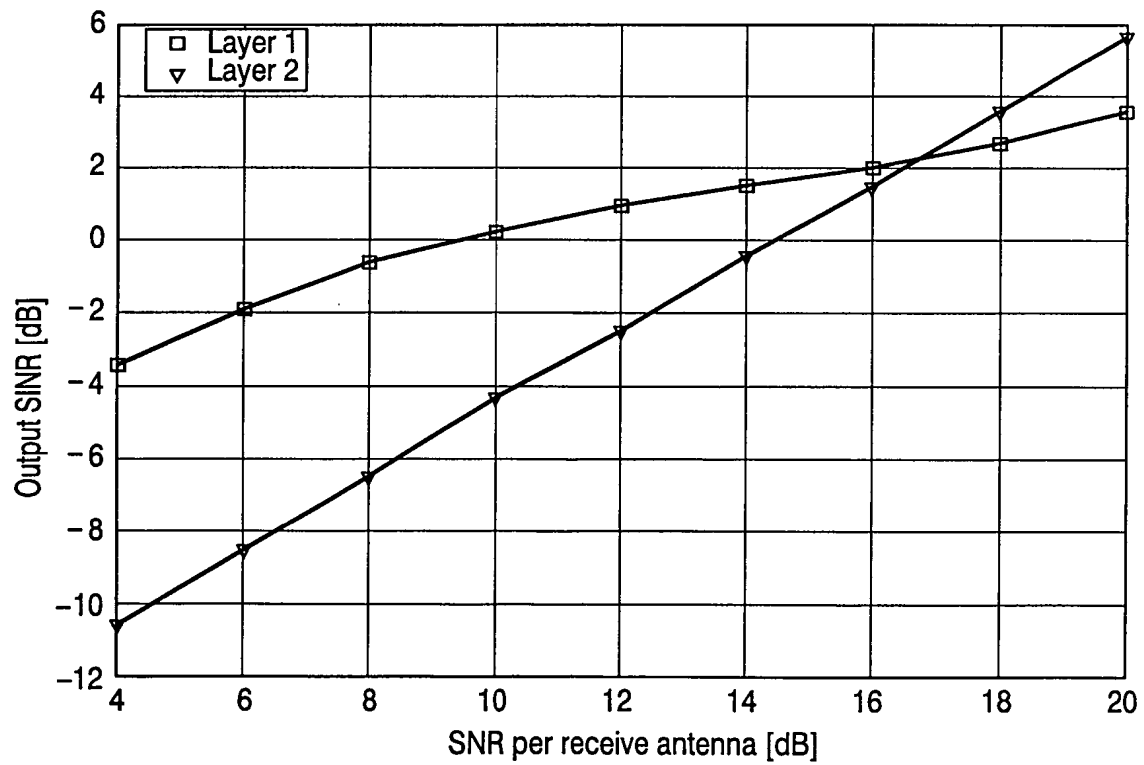
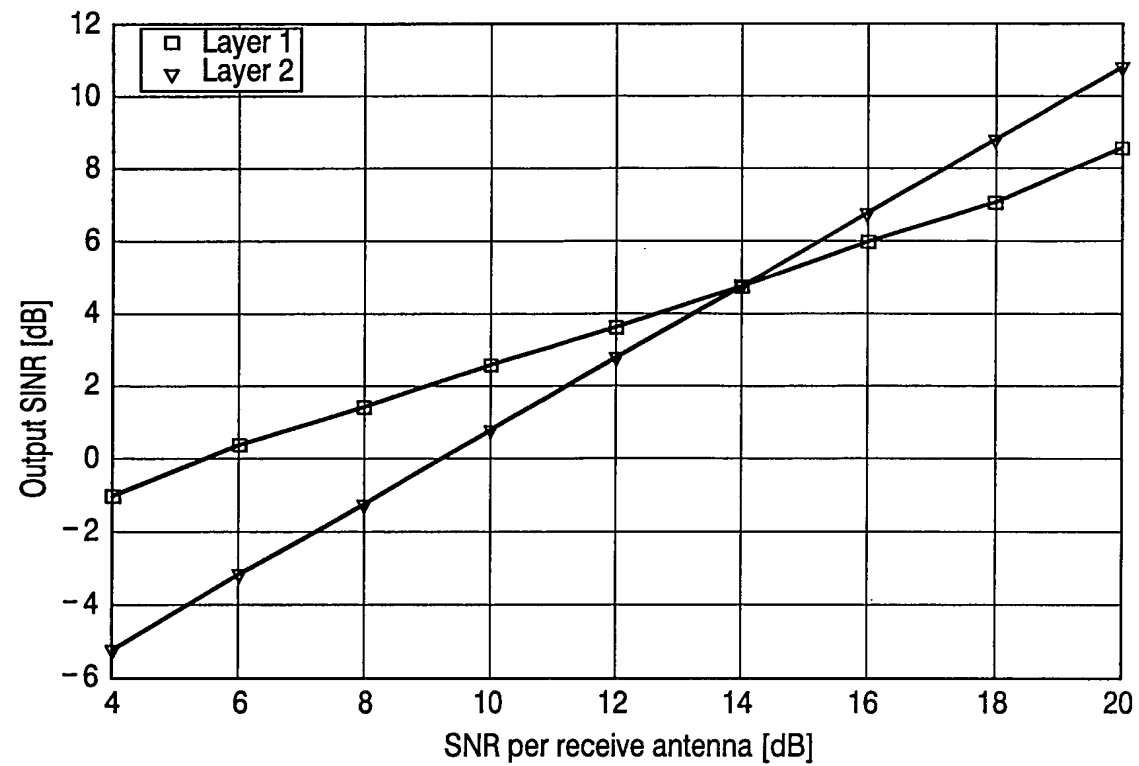


FIG. 8

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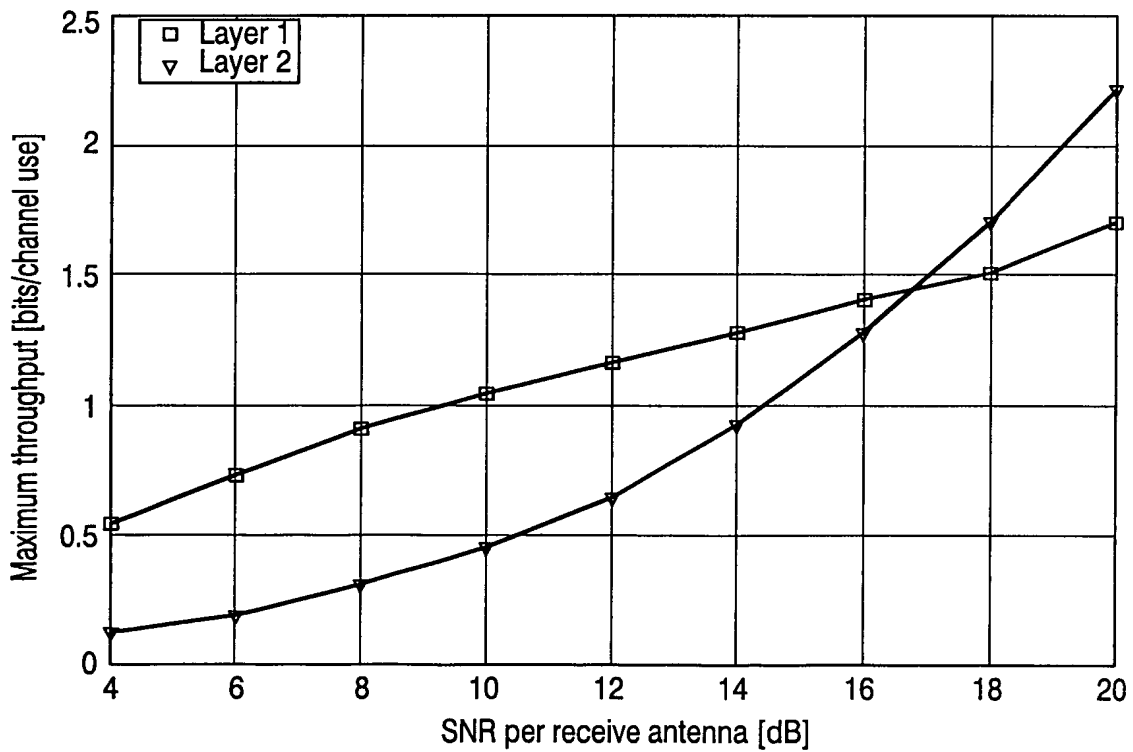
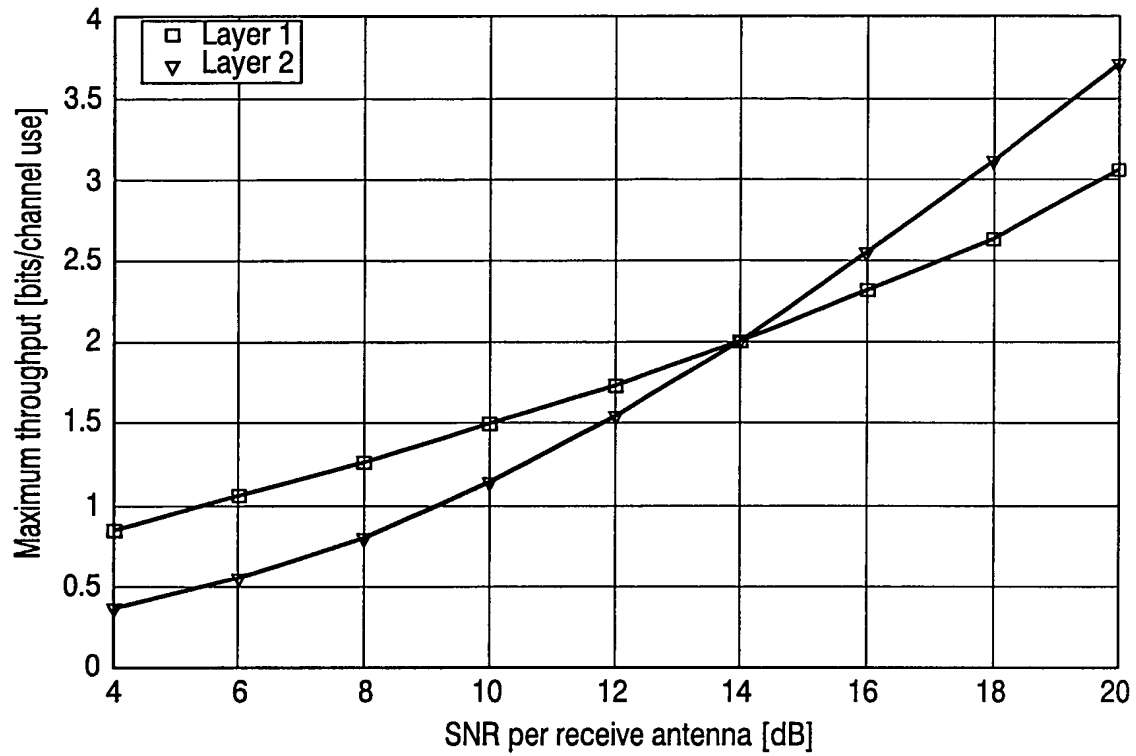


FIG. 9

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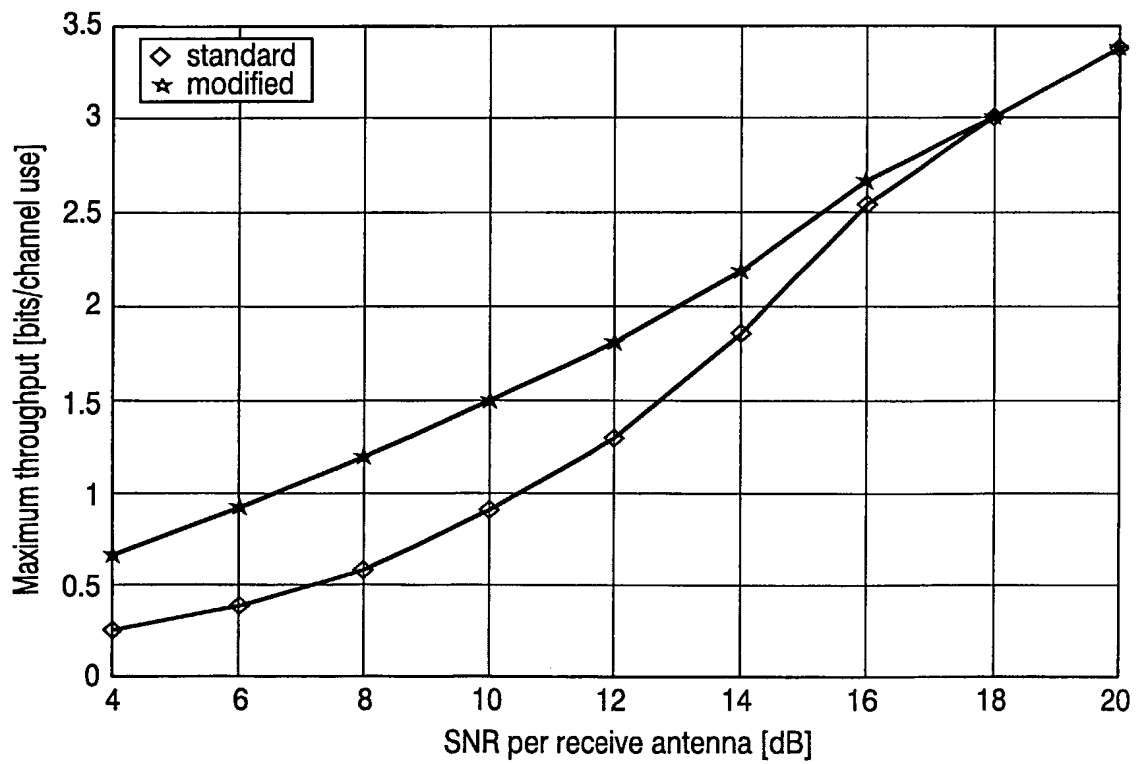
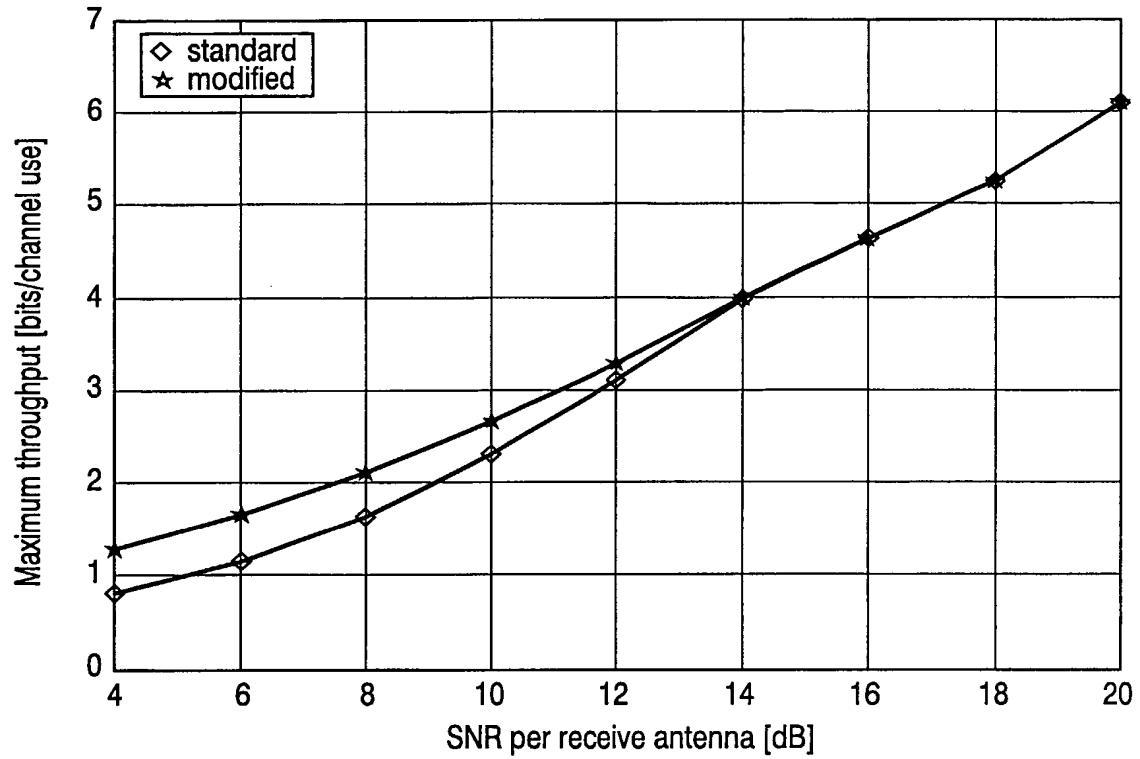


FIG. 10

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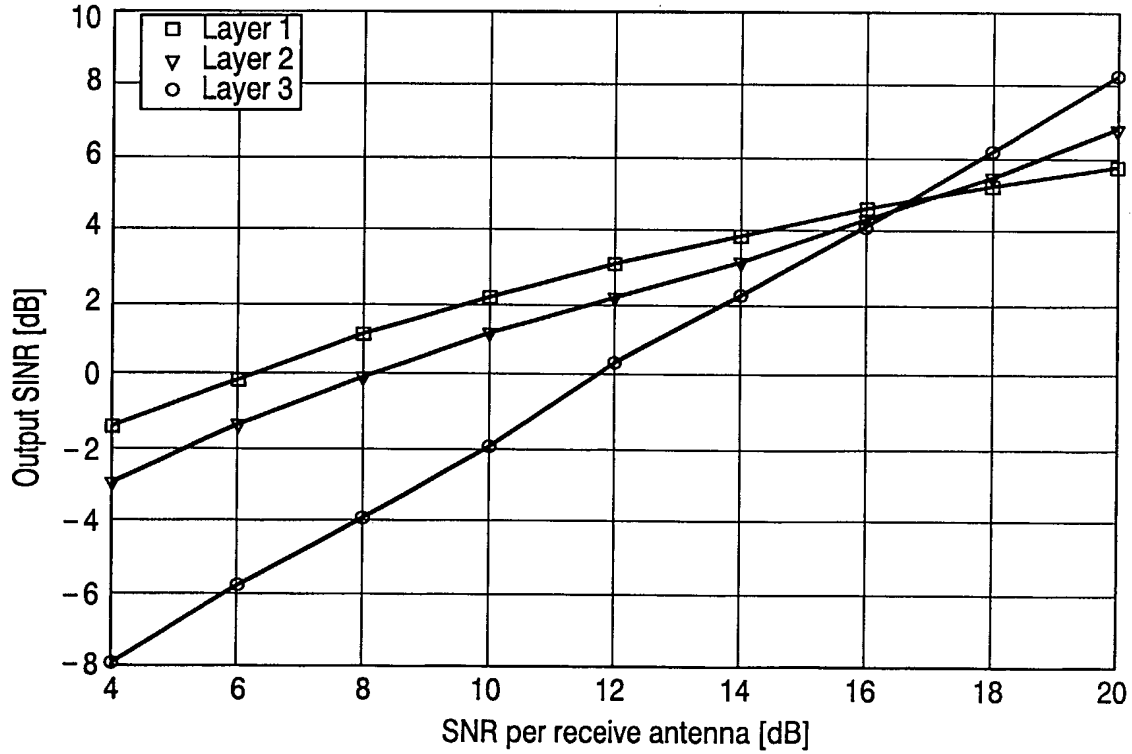
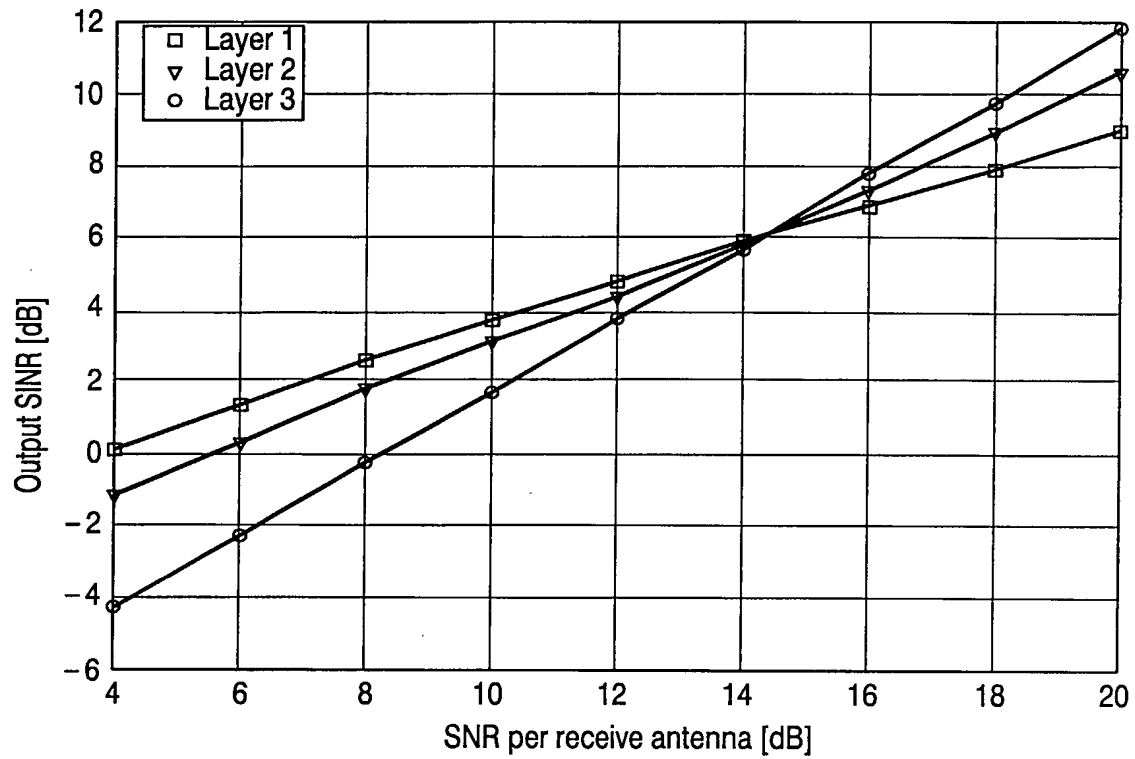


FIG. 11

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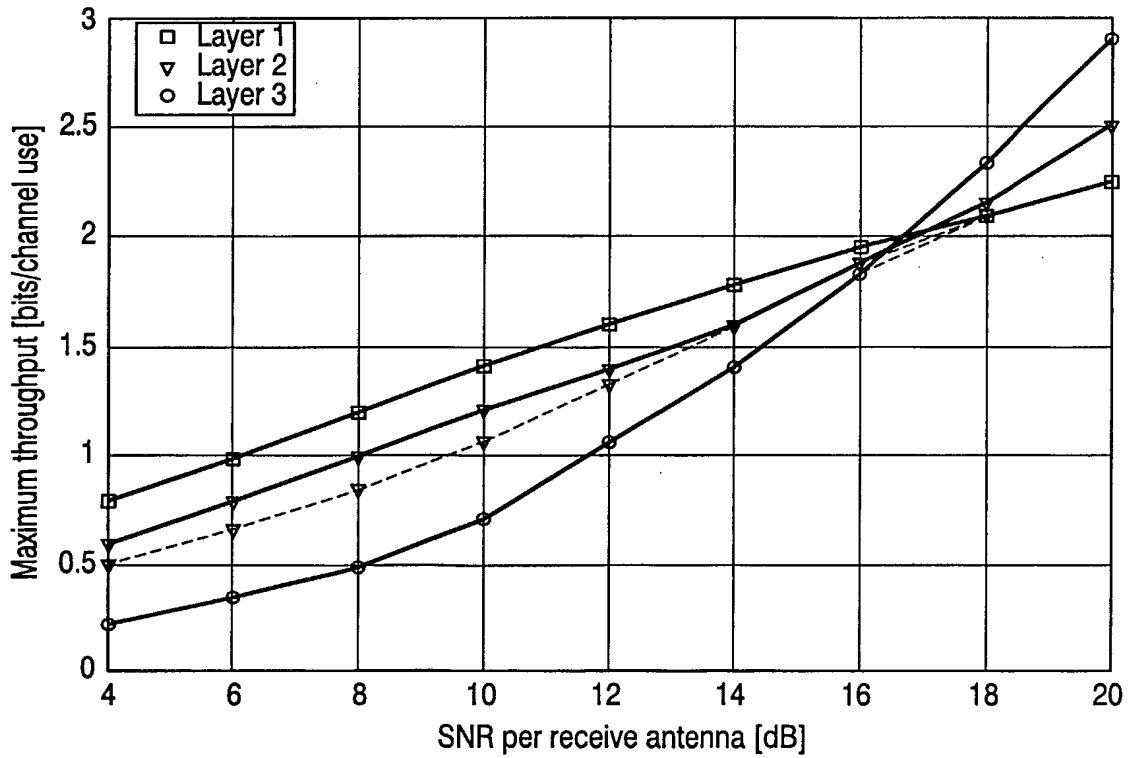
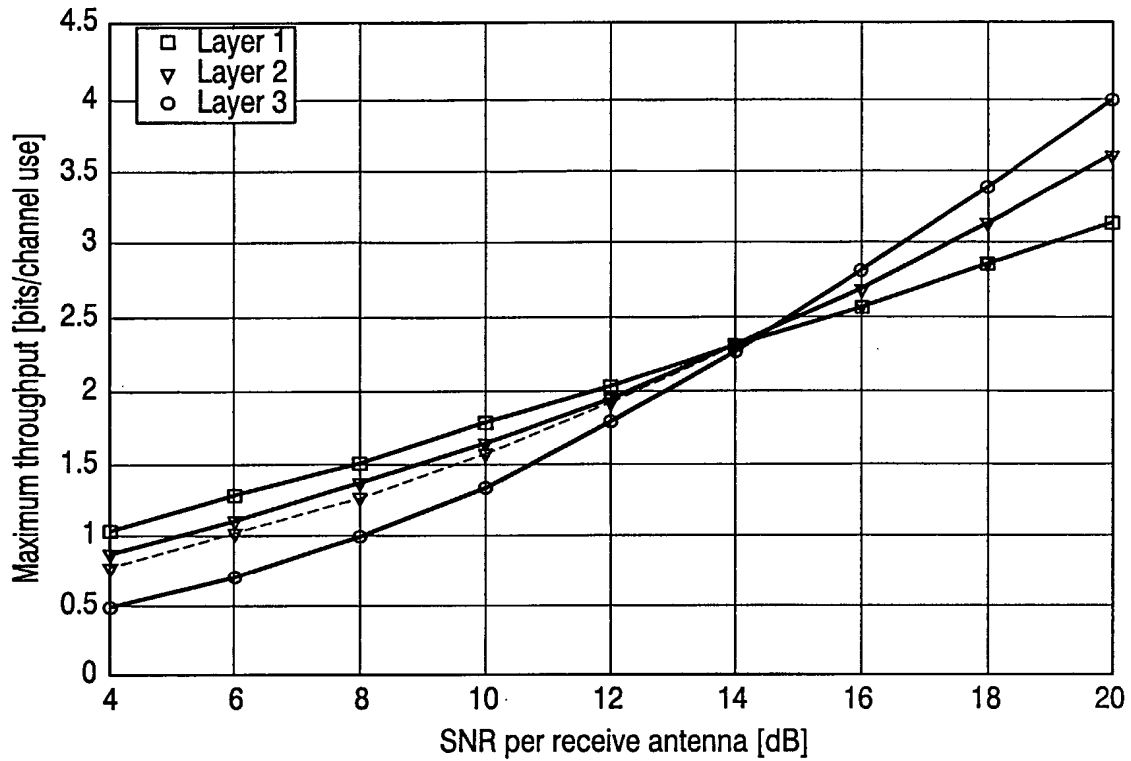


FIG. 12

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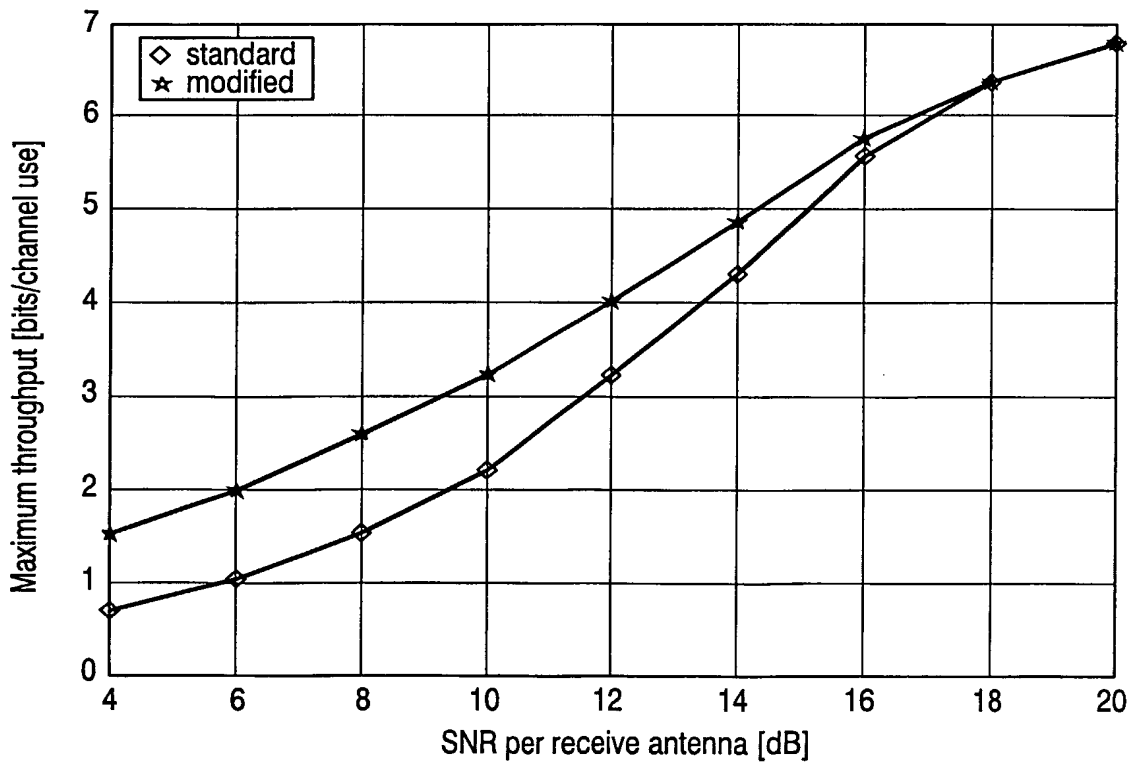
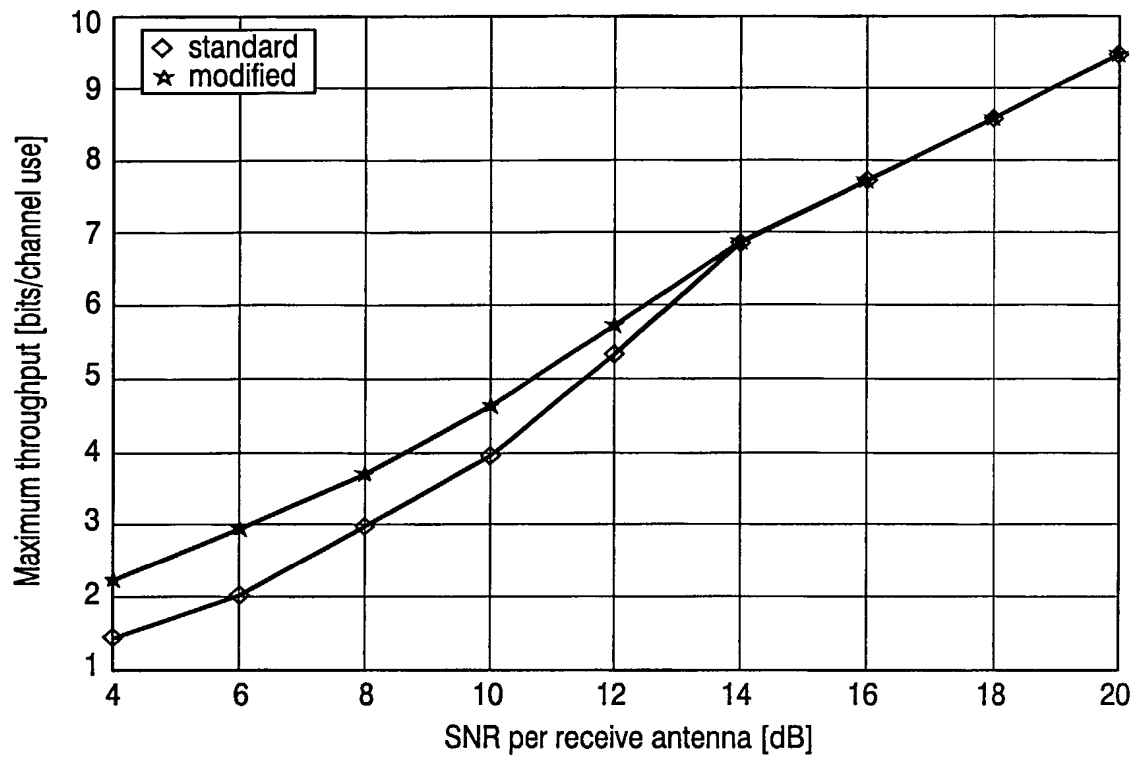


FIG. 13

INTERNATIONAL SEARCH REPORT

Int: Application No
PCT/IB 02/02290

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04L1/00 H04L1/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	CATREUX S ET AL: "SIMULATION RESULTS FOR AN INTERFERENCE-LIMITED MULTIPLE INPUT MULTIPLE OUTPUT CELLULAR SYSTEM" GLOBECOM'00. 2000 IEEE GLOBAL TELECOMMUNICATIONS CONFERENCE. SAN FRANCISCO, CA, NOV. 27 - DEC. 1, 2000, IEEE GLOBAL TELECOMMUNICATIONS CONFERENCE, NEW YORK, NY: IEEE, US, vol. 2 OF 4, 27 November 2000 (2000-11-27), pages 1094-1096, XP001017253 ISBN: 0-7803-6452-X the whole document	1-20
P,A	WO 01 69801 A (INTERDIGITAL TECH CORP) 20 September 2001 (2001-09-20) abstract figure 5	1-20



Further documents are listed in the continuation of box C.



Patent family members are listed in annex.

* Special categories of cited documents :

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Information on patent family members

Int. Application No

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